Lab 1: AM Systems

1 Overview

In this lab, we will deploy amplitude and frequency modulation systems using Pluto SDR. We will look into receiver models for AM (including double side band receivers) and attempt to receive a broadcast message from the transmitting node (based on a Pluto SDR). It is recommended to read up on AM systems include complex-envelope models, coherent and non-coherent detection (some details are provided in the appendix).

1.1 Outcomes

Upon successful completion of this laboratory, you should be able to:
- Observe transmission of information using amplitude/frequency modulation over a wireless channel.
- Understand modulation schemes such as DSBAM and DSB-SC waveforms based on time-domain and frequency-domain measurements.
- Configure a real-time DSBAM receiver using their Pluto SDR device.

1.2 Background information

Appendix A1 provide some background details and mathematical formulation relating to standard AM (also called double-sideband AM or DSBAM) and the suppressed carrier scheme DSB-SC (suppressed carrier).

2. AM Systems

First, we will use your individual Pluto SDR devices to observe amplitude modulated communication signals. For the purpose of this lab, we have a transmitter setup using a Pluto SDR that broadcast radio-frequency signals in the laboratory at a frequency in the 2.45 GHz frequency band (ISM band).

The broadcast waveform consists of both a DSBAM and a DSB-SC communication signal, separated from one another by 60 kHz (i.e., centre frequencies of the two transmissions are separated by 60 kHz). Both passband signals are the result of modulating a sinusoidal carrier with a sinusoidal message signal.

Our first step is to observe the transmitted signal and to observe its properties. At the second stage, you will build a receiver function to demodulate and decode the message.

2.1 Observe the AM signal

Using the Spectrum Analyser model you created in Lab 0, observe the communication signals in the frequency-domain. You should see a spectrum similar to Figure 1. Set the sampling rate to 240000 on your receiver to see the waveforms clearly. [NOTE: We may

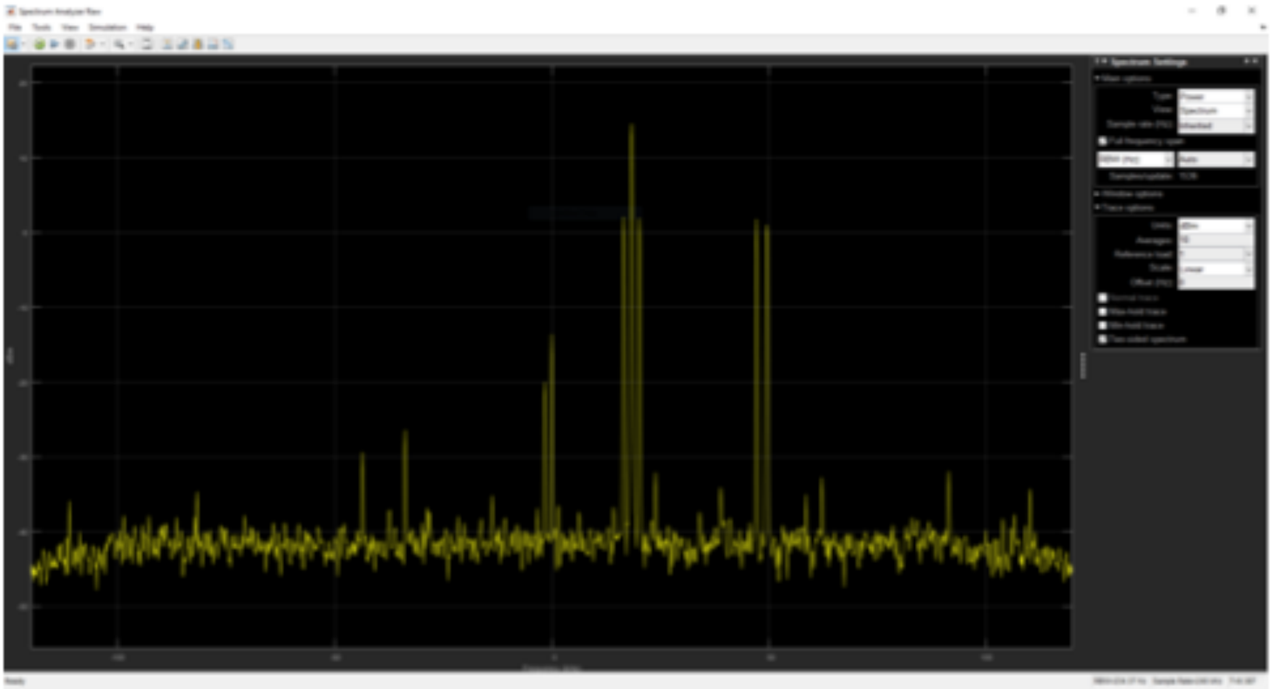


Fig 1: DSBAM and DSB-SC spectrum

set the Center Frequency to 2.49 GHz to minimise interference with 2.4 GHz wireless channels.]

Observe the waveform carefully to answer the following questions:

Question 1.1: On a screenshot of the spectrum analyser, identify the DSBAM, DSB-SC and their sidebands (you may annotate or define them in the text).
Question 1.2: Determine the actual carrier frequency of the DSBAM and the DSB-SC passband waveform?
Question 1.3: Can you determine the message frequency of the DSBAM communication signal?

2.2 Capturing in time-domain

The spectrum analyser captures and visualises the received signal in frequency domain. However, in some cases, a time-domain visualisation can be more useful to identify some parameters of the received signal. We can add the time domain visualisation to the Simulink model by using a filter and a time scope. The model should look similar to Figure 2 below. [NOTE: We may set the Center Frequency to 2.49 GHz to minimise interference with 2.4 GHz wireless channels.]

The Time Scope can be found under DSP System Toolbox → Sinks.

For the filter block, use DSP System Toolbox → Filtering → Filter Implementations → Digital Filter Design.

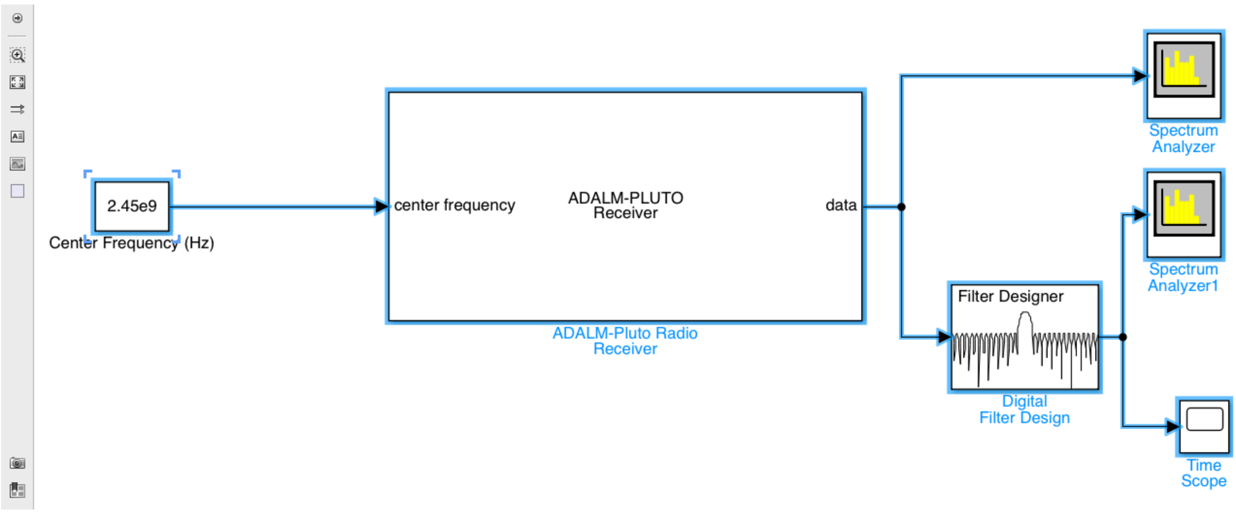


Fig 2: Time Domain Visualisation Simulink Model

The digital filter is used to isolate either the DSBAM or DSB-SC waveforms (based on your configuration). You should set it as a 60-order Equiripple band-pass filter surrounding one of the two transmissions (use the info from Question 2.2 above). [NOTE: If you do not isolate them, then the resulting time-domain plot will be a superposition of the two, which would be challenging to analyse].

You should modify the output of PLUTO device to be Single/Double precision floating point type instead of int16.

We also add a second spectrum analyser to validate that the isolation has been established by the bandpass filter.

On running the model, you should observe a time-domain waveform on the scope similar to below.

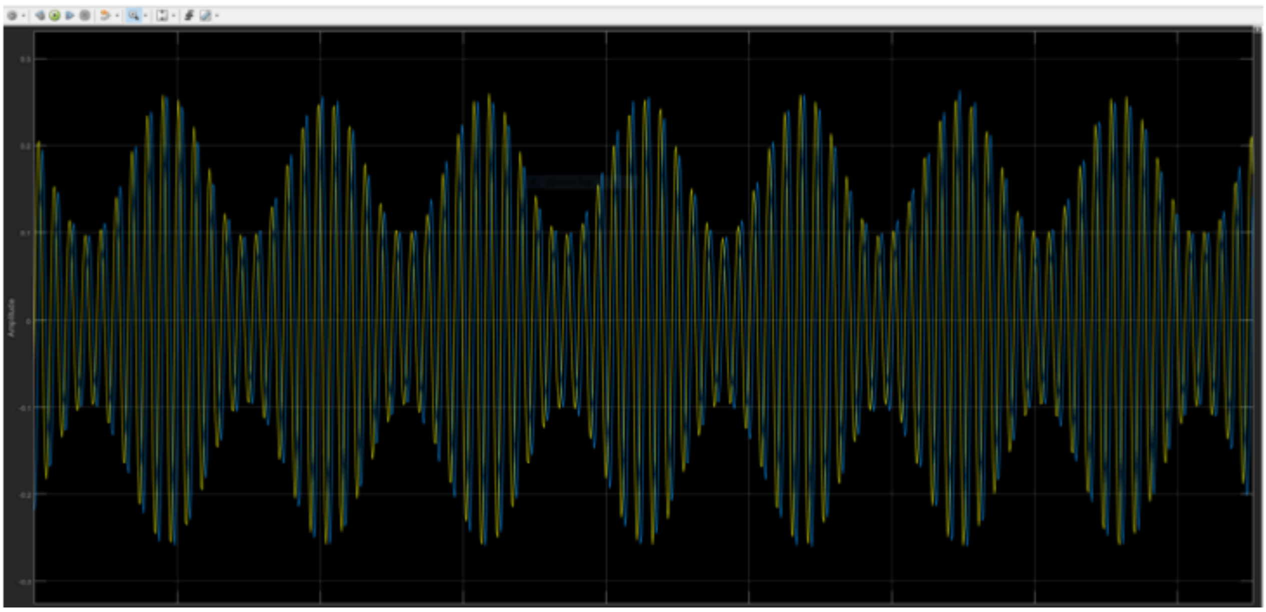


Fig 3: Time-domain signal from the Time Scope

Question 1.4: Configure the filters appropriately to see both the DSB-SC and DSBAM waveforms. Are you able to identify message frequency from the time-domain plots?

2.3 Complete DSBAM Receiver

Now that you have observed the waveforms and identified the parameters, we can construct the full processing chain for the DSBAM receiver. You will use this receiver to demodulate a voice signal that is being broadcast using the instructor's SDR platform. Though DSBAM voice communication has been around for many decades, it is still in use today in many application scenarios such as ATC ground communication (see here).

2.3.1 Theory: DSBAM Demodulation

Consider a DSBAM passband signal of the form

$$s_{DSBAM}(t) = [A + m(t)]\cos(2\pi f_c t)$$

where f_c is the carrier frequency, m(t) is the message signal, and A is chosen such that

$$A + m(t) > 0 \forall t$$

This choice of A means that the information signal m(t) is contained in the envelope of the passband signal.

Our demodulation model should:

- Tune the receiver's centre frequency near, but not exactly to the centre frequency f_c of the transmission. In many practical conditions, tuning the receiver to the exact central frequency is not feasible due to limitations of components (oscillator stability, time drift etc). Hence, we will tune the receiver to a frequency $f_c - f_i$, with $f_i \ll f_c$. The offset f_i must be small enough that the DSBAM signal remains within the instantaneous bandwidth of the SDR receiver.

- Filter the received signal to limit noise. Subsequently, extracting the envelope of the received DSBAM waveform to detect the message. This is a form of non-coherent demodulation.

Theoretical background is covered in more detail in Appendix A1. In short, assuming that the receiver generates samples of $\hat{z}(t)$, the message signal can be recovered through these steps as seen below.

$$z(t) = \text{LPF}\left[s_{DSBAM}(t)\, e^{-j(2\pi(f_c - f_i)t + \varphi)}\right] = \frac{1}{2}[A + m(t)]\, e^{j(2\pi f_i t - \phi)}$$

and further by computing the magnitude of this signal,

$$\hat{m}(t) = |z(t)| = \frac{1}{2}[A + m(t)]$$

arriving at our original message offset by a DC value.

The important observation is that despite the frequency offset f_i and phase offset ϕ, we arrive at our original message (though offset by a DC value).

2.3.2 Creating the Simulink model

Expand the Simulink model from before to incorporate the following changes:

- Modify the PLUTO-SDR Receiver block's configuration:
  - Sampling rate: 240e3
  - Output data type: single
  - Samples per frame: 1000
  - Center frequency and Gain: set to input port

- Add a *constant block* and *slider gain* block to set a tunable gain for the receiver. Set the max gain value to 65 and minimum to 0, allowing the gain to be varied between these levels.
- Add more *spectrum analyzer* and *time scope* blocks to debug the model at different stages and to view the signals of interest. Once validated, they can be removed to achieve real-time performance

[**NOTE**: The performance achieved by these blocks is dependent on the capabilities of your machine. Since most of the Simulink blocks process data from the ADALM on your machine, adding more visualisations and debugging tools hinder the forward path performance. In practice, some of these blocks could be implemented on the ADALM hardware, but this requires the full design flow from Simulink design to HDL to integration with Linux OS on the ADALM board]

- Modify the bandpass filter block, if needed, to the frequency of the DSBAM carrier.

- Add a *Complex to Magnitude-Angle* block from Simulink → Math Operations. This block will demodulate the DSBAM signal at f_i to 0 Hz.

- Add an *FIR Decimation* block from DSP System Toolbox → Filtering → Multirate Filters. This block changes the sample rate of the signal based on the decimation factor specified in its configuration. Choose a decimation factor of 5, reducing the sample rate of the signal by a factor of 5 at the output of the block. This is to bring the sampling rate at the audio (message) stage to (240 kHz)/5 = 48 kHz, which is a standard sample rate for audio signals. [NOTE: you might need to change this to meet the audio sample specification for accurate reproduction]

- To remove the DC offset, add a *DC Blocker* block from DSP System Toolbox → Signal Operations (to remove the offset A in our case).

- Finally, add an Audio Device Writer block from DSP System Toolbox → Sinks to interface to your PC sound card. Leave this block at default settings as it should work for most desktop/laptops.

The final model can be seen in Figure 4 below:

- Now set the simulation stop time to *inf* and run the model.

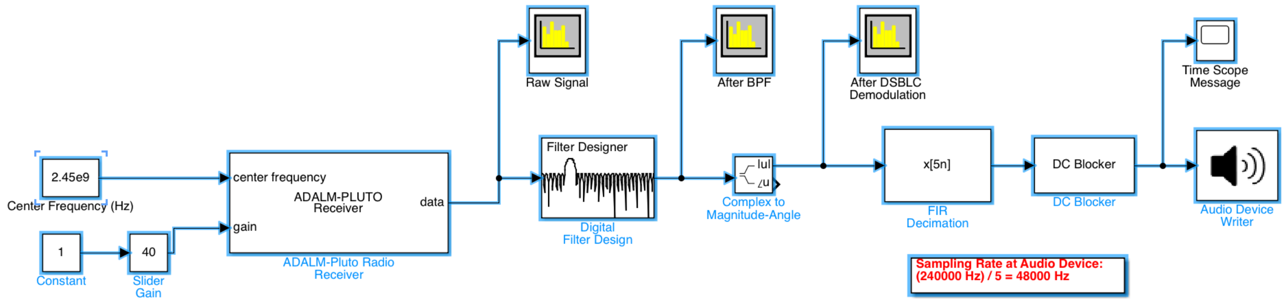


Fig 4: Simulink model for DSBAM receiver

- You may have to adjust the center frequency according to the design of your bandpass filter and viceversa to get the system to function.
- Check the Spectrum Analyzer window(s). Are you seeing what is expected (based on the theory in 2.3.1)?
- Are you able to hear the audio sample? [you might need to tweak the receiver gain setting to improve the reception].

Question 1.5: What happens to the recovered audio when you adjust the gain of the SDR receiver block? Summarise your findings.

Question 1.6: What happens when you remove the Bandpass Filter from the model? [You can do this easily by right-clicking the block and selecting “Comment Through”]. Summarise your findings.

Submission

Submit the following for your observation as a short report via blackboard:

- Screen shot and notes from AM observations
- Screen capture of the time-domain signal
- Observations from your DSBAM receiver model (spectrum analyser captures)
- Thoughts on question 1.1 through 1.6

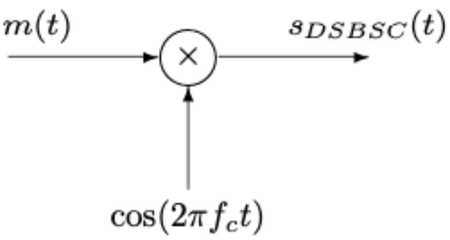
Acknowledgement: Labs adapted from C. Prust, Introductory Communication Systems Course using SDR, 2022.

Appendix A1:

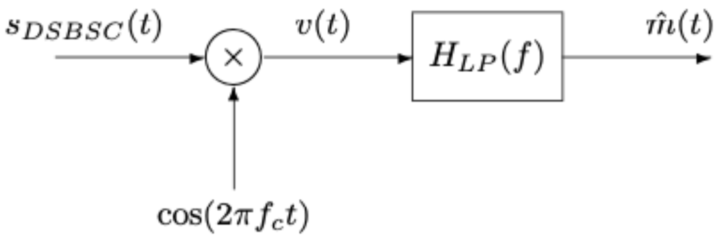
Double-Sideband Suppressed Carrier (DSB-SC)

DSB-SC is an AM format where the modulated signals are symmetrically spaced above and below the carrier frequency, while the carrier level is (ideally) completely suppressed (see the Wiki page here)

The simplified modulator can be visualised as shown below, where $m(t)$ is the message signal and f_c is the carrier frequency. The demodulator uses the inverse operation and a low pass filter $H_{LP}(f)$ to determine the recovered message $\hat{m}(t)$.



DSB-SC modulator



DSB-SC demodulator

In time domain, these operations can be represented using the equations below.

Modulation: $s_{DSBSC}(t) = m(t)\ \cos(2\pi f_c t)$

Demodulation: (LPF = low pass filter)

$$\begin{aligned}\hat{m}(t) &= \text{LPF}\left[s_{DSBSC}(t)\ \cos(2\pi f_c t)\right] \\ &= \text{LPF}\left[m(t)\ \cos(2\pi f_c t)\ \cos(2\pi f_c t)\right] \\ &= \text{LPF}\left[m(t)\{1 + cos(2\pi(2f_c)t\}\right] \\ &= \frac{1}{2}m(t)\end{aligned}$$

The modulation operation is more evident in the frequency domain:

$$\begin{aligned}S_{DSBSC}(f) = \mathscr{F}\{m(t)\ \cos(2\pi f_c t)\} &= \mathscr{F}\{m(t)\} * \mathscr{F}\{\cos(2\pi f_c t)\} \\ &= M(f) \times \frac{1}{2}\{\delta(f - f_c) + \delta(f + f_c)\} \\ &= \frac{1}{2}M(f - f_c) + \frac{1}{2}M(f + f_c)\end{aligned}$$

i.e., the message are symmetrically spaced around the carrier signal frequency.

In case of demodulation,

$$\begin{aligned} V(f) &= S_{DSBSC}(f) \times \mathscr{F}\{\cos(2\pi f_c t)\} \\ &= \frac{1}{2}\left[\Big(M(f-f_c) + M(f+f_c)\Big) \times \Big(\delta(f-f_c) + \delta(f+f_c)\Big)\right] \\ &= \frac{1}{2}M(f) + \frac{1}{4}M(f+2f_c) + \frac{1}{4}M(f-2f_c) \end{aligned}$$

$$\implies \hat{M}(f) = H_{LP}(f) \times V(f) = \frac{1}{2}M(f)$$

where, $H_{LP}(f)$ is the frequency response of the low-pass filter.

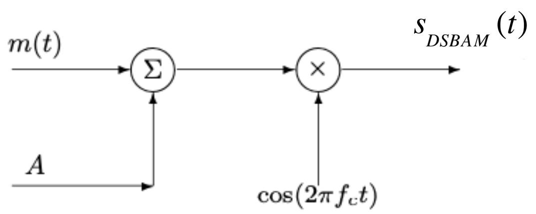
Double-Sideband Amplitude Modulation (DSBAM)

Amplitude modulation is a modulation technique where the amplitude or strength of the carrier oscillations is varied in accordance with the modulating signal's amplitude. A common example is the AM radio communication where a continuous wave radio-frequency signal (typically a sinusoidal carrier wave) has its amplitude modulated by an audio waveform that needs to be transmitted. The audio waveform modifies the amplitude of the carrier wave and determines the envelope of the waveform.

This operation, when viewed in the frequency domain, results in a signal with power concentrated at the carrier frequency and two adjacent sidebands. Each sideband is equal in bandwidth to that of the modulating signal, and is a mirror image of the other. Hence, standard AM is also referred to as DSBAM.

Note that the carrier does not convey any information to the receiver and is thus wasting energy by being transmitted, resulting in other AM formats like DSB-SC, DSB-RC (double-sideband reduced carrier) and SSB (single-sideband) transmissions. (see wiki page here)

A standard AM modulator can be visualised using the operation shown below, with the symbols having the standard meaning.



The modulation operation in time-domain is

$$s_{DSBAM}(t) = [m(t) + A]\,\cos(2\pi f_c t)$$

and in the frequency-domain, we get

$$S_{DSBAM} = \mathscr{F}\left\{[m(t)+A]\cos(2\pi f_c t)\right\}$$
$$= [M(f)+A\times\delta(f)]\times\frac{1}{2}\left(\delta(f-f_c)+\delta(f+f_c)\right)$$
$$= \frac{1}{2}\left[A\times\delta(f-f_c)+M(f-f_c)+A\times\delta(f+f_c)+M(f+f_c)\right]$$

In case of the complex signal model for the SDR receiver, the samples of $\hat{z}(t)$ are generated by the receiver from the RF signal. Using the DSBAM modulator model with an offset frequency f_i, we know that

$$z(t) = \text{LPF}\{s_{DSBAM}(t)\,e^{-j(2\pi(f_c-fi)t+\phi)}\}$$
$$= \text{LPF}\{[A+m(t)]\,e^{-j(2\pi(f_c-fi)t+\phi)}\}$$
$$= \frac{1}{2}\{[A+m(t)]\left(e^{j(2\pi f_i t-\phi)}+e^{-j(2\pi(2f_c+f_i)t-\phi)}\right)\}$$
$$= \frac{1}{2}[A+m(t)]e^{j(2\pi f_i t-phi)}$$

Here, the low pass filter removes the double-frequency term near $2f_c$ since we have defined the offset f_i as $f_i \ll f_c$.
Thus our receiver can compute the magnitude of the above signal to determine the original message offset by a constant dc amplitude. The operation in frequency domain is capture in the figure below.